IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

#10
m
3·20·03

Applicant: Oh et al
Serial No.: 09/483,569
Filed: January 14, 2000
For: SIMPLIFIED NOISE SUPPRESSION CIRCUIT

Art Unit: 2654
Examiner: Talivaldis Smits
Docket: TI-23373

OFFICIAL
RECEIVED
3/18/03

Appeal Brief under 37 C.F.R. §1.192

Assistant Commissioner
for Patents
Washington, D.C. 20231

CERTIFICATION OF FAX TRANSMITTAL

I hereby certify that the above correspondence is being facsimile transmitted to the Patent and Trademark Office on **March 18, 2003.**

Robin Barnum
Robin E. Barnum

Dear Sir:

This is Appellant's Appeal Brief filed pursuant to 37 C.F.R. §1.192 and the Notice of Appeal filed January 22, 2003.

Real Party in Interest under 36 C.F.R. §1.192(c)(1)

The real party in interest in this application is Texas Instruments Incorporated, a corporation of Delaware with its principal place of business in Dallas, Texas. An assignment to Texas Instruments Incorporated is recorded at reel 010513 and frames 0488 to 0490.

Related Appeals and Interferences under 36 C.F.R. §1.192(c)(2)

03/21/2003 SMOORE 00000002 200668 09483569

There are no other appeals or interferences related to this appeal in this application.

01 FC:1402 320.00 CH

Status of the Claims on Appeal under 37 C.F.R. §1.192(c)(3)

Claims 1 to 3 and 9 to 11 are finally rejected. Claims 4 to 8 and 12 to 22 are canceled. No claims are allowed.

Status of Amendments Filed After Final Rejection under 37 C.F.R. §1.192(c)(4)

The response after final rejection filed December 16, 2002 canceled claims 6, 14 and 17 to 22. The ADVISORY ACTION of January 6, 2003 stated that this amendment would be entered upon filing this appeal.

Summary of the Invention under 37 C.F.R. §1.192(c)(5)

This invention is a method and apparatus for reducing noise in a sampled acoustic signal. A sampler obtains discrete samples of an acoustic signal. An analog to digital converter forms a stream of sampled acoustic signals. The invention selects a fixed number of samples. This fixed number of samples is preferably 32 samples. The invention multiplies these samples by a windowing function. This windowing function is preferably a hanning window function. A fast Fourier transform of the windowed samples yields transformed windowed signals. The invention selects half of the transformed windowed signals. The invention calculates a power estimate of the transformed windowed signals and a smoothed power estimate by smoothing the power estimate over time. The invention calculates a noise estimate. Then invention calculates a gain function from the noise estimate and the smoothed power estimate. The invention calculates a transformed speech signal by multiplying the gain function with the transformed windowed signal. An inversed fast Fourier transform of the transformed speech signal yields a sampled speech signal. The invention adds the sampled speech signal to a portion of the speech signal of a previous frame.

Received from < 972 917 4418 > at 3/18/03 5:25:46 PM [Eastern Standard Time]

Statement of Issues Presented for Review under 37 C.F.R. §1.192(c)(6)

Are claims 1 to 3 and 9 to 11 made obvious under 35 U.S.C. 103(a) by Bloebaum et al, U.S. Patent No. 6,070,137?

Statement of the Grouping of Claims under 37 C.F.R. §1.192(c)(7)

The Applicants respectfully submit that the claims of this application are independently patentable in the following groups:

Group I; claims 1 and 9, claims 2 and 3 stand with claim 1 and claims 10 and 11 stand with claim 9.

This Appeal Brief includes separate arguments for each of these groups. In accordance with the procedure sanctioned in MPEP §1206(5) the Appellant respectfully submits these separate arguments fulfill the requirement of 37 C.F.R. §1.192(c)(6) for statement of the reason why the claims are believed separately patentable.

Arguments

Group I

Claims 1 to 3 and 9 to 11 were finally rejected under 35 U.S.C. 103(a) as made obvious by S. Bloebaum et al. U.S. Patent 6,070,137, filed January 7, 1998.

Claims 1 and 9 recite subject matter not made obvious by Bloebaum et al. Claim 1 recites "calculating a smoothed power estimate by smoothing the power estimate over time." Likewise, claim 9 recites the noise suppression circuit operates to "calculate a power estimate of the transformed windowed signals." The FINAL REJECTION states at page 5, lines 12 and 13 that Bloebaum et al teaches:

"smoothing the power estimate over time when there is no speech to calculate a noise power estimate (col. 5, lines 37-44 and 60-65)"

Bloebaum et al states at column 5, lines 30 to 44:

"The adaptation process involves smoothing of the model parameters in order to reduce the variance of the noise estimate. This may be done using either a moving average (MA), autoregressive (AR), or a combination ARMA process. AR smoothing is the preferred technique, since it provides good smoothing for a low ordered filter. This reduces the memory storage requirements for the noise suppression algorithm. The noise model adaptation with first order AR smoothing is given by the following equation:

$$N^{(i)} = \alpha N^{(i-1)} + (1-\alpha) S,$$

where α may be in the range $0 \leq \alpha \leq 1$, but is further constrained to the range $0.8 \leq \alpha \leq 0.95$ in the preferred embodiment of the invention."

This portion of Bloebaum et al clearly teaches smoothing of the vector N from noise model adaption block 46 as a function of the prior noise vector N and the vector S. This is not smoothing the power estimate as claimed.

Claims 1 and 9 each recite calculation of "a gain function from the noise estimate and the smoothed power estimate." Bloebaum et al illustrates transform and filter computation block 56 which receives the power spectral density (PSD) estimate represented by $|S(e^{j\omega})|^2$ from block 44 and the vector N from noise model adaption block 46 and produces enhancement filter $|H(e^{j\omega})|$. If the vector N is the claimed smoothed power estimate, then transform and filter computation block 56 receives the power spectral density estimate from block 44 and the smoothed power spectral density estimate (vector N) from noise model adaption block 46. These are not the inputs to the calculated gain function recited in claims 1 and 9.

Received from < 972 917 4418 > at 3/18/03 5:25:46 PM [Eastern Standard Time]

Thus if the vector N is the claimed smoothed power estimate, Bloebaum et al fails to make obvious a different limitation of claims 1 and 9. Accordingly, the Appellants respectfully submit that claims 1 and 9 are allowable over Bloebaum et al.

Claims 2, 3, 10 and 11 are allowable by dependency upon allowable base claims.

If the Examiner has any questions or other correspondence regarding this application, Applicants request that the Examiner contact Applicants' attorney at the below listed telephone number and address to facilitate prosecution.

Texas Instruments Incorporated
P.O. Box 655474 M/S 3999
Dallas, Texas 75265
(972) 917-5290
Fax: (972) 917-4418

Respectfully submitted,

Robert D. Marshall, Jr.

Robert D. Marshall, Jr.
Reg. No. 28,527

APPENDIX

CLAIMS ON APPEAL

1. A method for reducing noise in a sampled acoustic signal, comprising:

receiving a stream of sampled acoustic signals;

selecting a fixed number of samples;

multiplying the samples by a windowing function;

computing the fast Fourier transform of the windowed samples to yield transformed windowed signals;

selecting half of the transformed windowed signals;

calculating a power estimate of the transformed windowed signals;

calculating a smoothed power estimate by smoothing the power estimate over time;

calculating a noise estimate;

calculating a gain function from the noise estimate and the smoothed power estimate.

calculating a transformed speech signal by multiplying the gain function with the transformed windowed signal;

calculating an inversed fast Fourier transform of the transformed speech signal to yield a sampled speech signal; and

adding the sampled speech signal to a portion of the speech signal of a previous frame.

2. The method of Claim 1, wherein the fixed number of samples is thirty-two.

3. The method of Claim 1, wherein the windowing function is a hanning window function.

9. A system for reducing noise in an acoustical signal comprising:

a sampler for obtaining discrete samples of the acoustical signal;

an analog to digital converter coupled to the sampler an operable to convert the analog discrete samples into a digitized sample;

a noise suppression circuit coupled to the analog to digital converter and operable to:

receive the analog discrete samples;

select a fixed number of samples;

multiply the samples by a windowing function;

compute the fast Fourier transform of the windowed samples to yield transformed windowed signals;

select half of the transformed windowed signals;

calculate a power estimate of the transformed windowed signals;

calculate a smoothed power estimate by smoothing the power estimate over time;

calculate a noise estimate;

calculate a gain function from the noise estimate and the smoothed power estimate.

calculate a transformed speech signal by multiplying the gain function with the transformed windowed signal;

calculate an inversed fast Fourier transform of the transformed speech signal to yield a sampled speech signal; and

add the sampled speech signal to a portion of the speech signal of a previous frame.

10. The system of Claim 9, wherein the fixed number of samples is thirty-two.

11. The system of Claim 9, wherein the windowing function is a hanning window function.

Received from < 972 917 4418 > at 3/18/03 5:25:46 PM [Eastern Standard Time]